A Medical Device to Detect Sound Sensitivity in Patients

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Introduction

For our capstone project, my team developed a system to play audio tones through a set of loudspeakers and measure a test subject's vital sign response. The properties of a sound can vary widely depending on the type of loudspeakers and the room in which it is played, and for our system to work as intended, we needed precise control over the sound reaching the patient's ears. To do this, we researched methods to calibrate our audio system to the acoustics of the test environment and loudspeakers. This report examines various solutions to this problem, including designing an acoustically treated testing environment and calibrating the loudspeakers for flat frequency response. All methods presented in this report are theoretical, as we did not have time to implement them into our system.

Acoustics: A Background

Every sound, from the popping of a balloon to a piece of music, can be decomposed as a sum of pure tones at different pitches. These tones (or frequencies, measured in Hz) make up a sound's *frequency spectrum*, which is a measure of the loudness present at each frequency. When passed through a *system* such as a loudspeaker, guitar amplifier, or a concert hall, each frequency reacts differently and can become louder or quieter relative to the others. The overall description of how a system affects a sound's spectral content is called the *frequency response* of the system.

In the physical world, sound exists as pressure waves in which the density of air varies over time and space. These waves emanate from a source, such as a loudspeaker, and reflect off ceilings, walls, and the floor before reaching the listener's ears (Aslan and Paurobally 608). The field of acoustics examines how the shape and material of surfaces in a room change the spectral content of a sound. When testing our capstone project on a patient, we discovered that the acoustics of our test environment (the patient's living room) created unwanted, uncharacterized variations in our sounds' frequency spectrum. Our overall goal, therefore, was to "flatten" the response of our audio system so the frequency spectrum of every sound was the same at the source (an audio file) and at the patient's ear.

Methods of Acoustic Treatment

Acousticians employ many techniques to flatten frequency response of a room. But to understand how to fix spectral unevenness, we must first understand its causes.

Causes of uneven frequency response

There are two main causes of uneven frequency response in an acoustical system: early reflections and bass response. Early reflections are delayed copies of a sound that reach the ear after bouncing off a nearby surface, as shown in Figure 1. Note that for the scope of this project, we disregarded reverberations which occur when there are many long-decaying reflections. Early reflections change the loudness of certain frequencies and harm the overall frequency response (Winer 447).

Another common issue in a listening environment is the existence of bass modes and nulls, which are peaks and valleys in the low-frequency response located at halfway points in a room (Winer 446). These two issues often contribute to unevenness in frequency response across the audio spectrum.

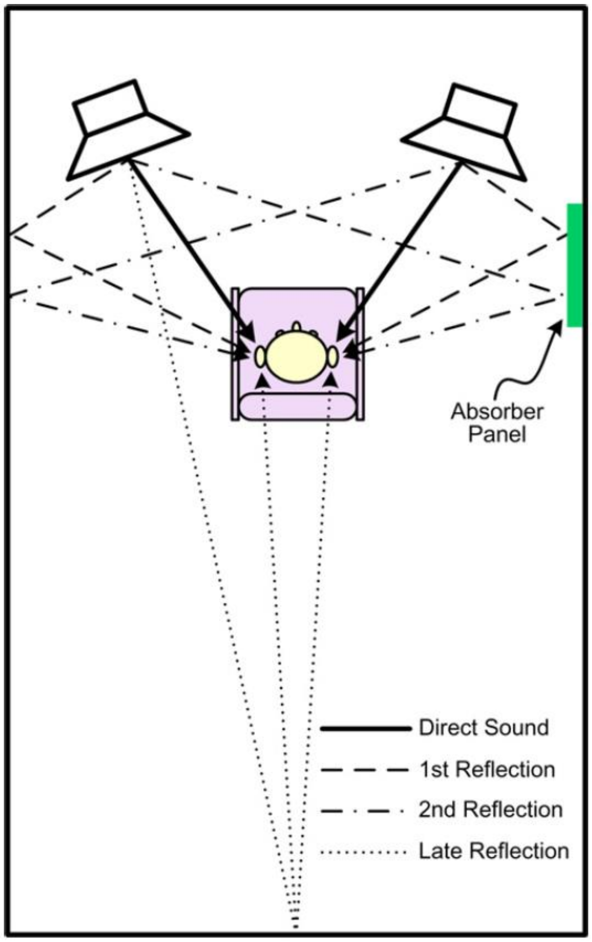

Figure 1: Illustration of early reflections (Winer 451)

Designing the test environment

Many studio recording engineers design their listening environments to eliminate bass nulls/modes and early reflections. We borrowed their techniques and practices to design an idealized test environment.

To create an optimal test environment, we must position the speakers and listener such that bass nulls and modes are reduced. To achieve this, the speakers should fire the long way through the room, and the listener shall be positioned approximately 38% away from the wall to create an equilateral triangle with the speakers (Winer 442-444). Both the speakers and the listener should be positioned slightly off-center, so the listener does not sit in a null or mode (Winer 446). In addition, the subject should sit with their ears at the same height as the speakers' high-frequency tweeters (Winer 444). To eliminate first and second reflections, we ornament the side walls with sound-absorbent panels whose positions can be calculated analytically or determined experimentally by use of a mirror (Winer 455). The diagram in Figure 2, borrowed from Winer, depicts our ideal testing environment.

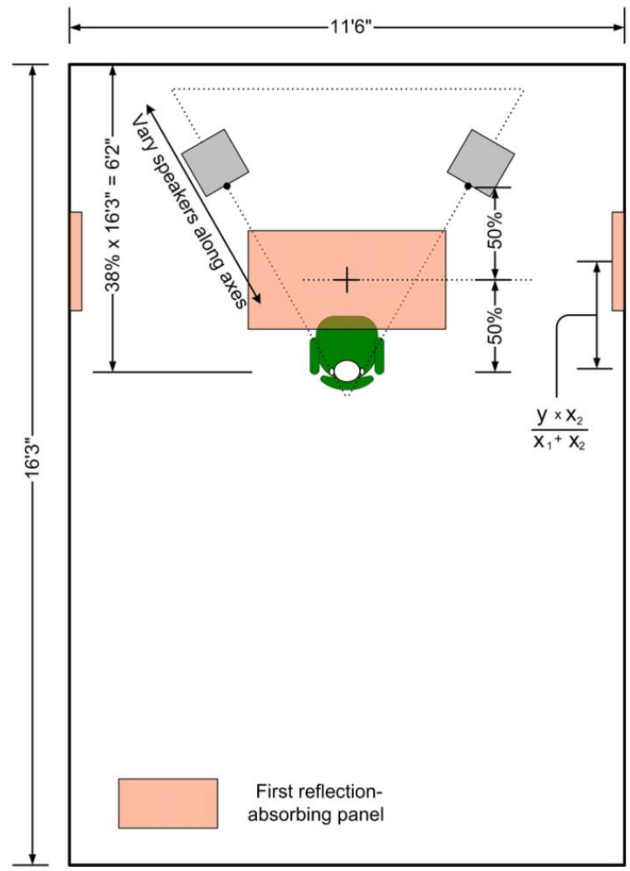

Figure 2: An ideal listening environment (Winer 444)

Figure 3 shows the effects of adding absorber panels to eliminate early reflections. Notice that the peaks and valleys are much less pronounced, as desired.

Because bass frequencies have longer wavelengths, they are more difficult to tame and thus require more substantial absorption. To flatten the bass response of our room, we employ additional "bass trap" absorbers in the corners where bass frequencies gather (Winer 458). Figure 4 below shows the frequency response of Winer's studio before and after adding bass traps.

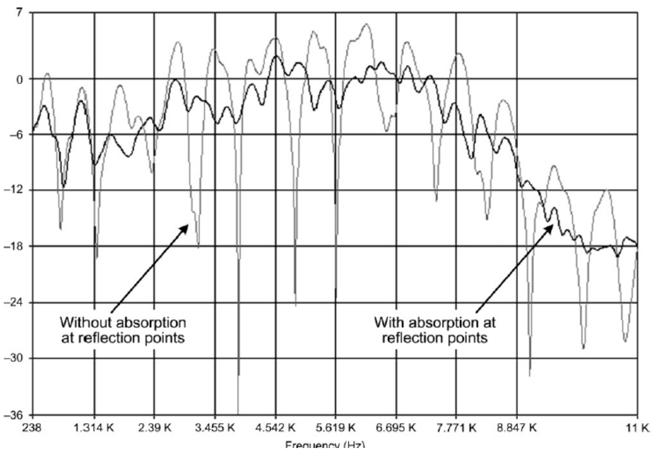

Figure 3: Adding absorptive panels at reflection points flattens the overall response (Winer 448).

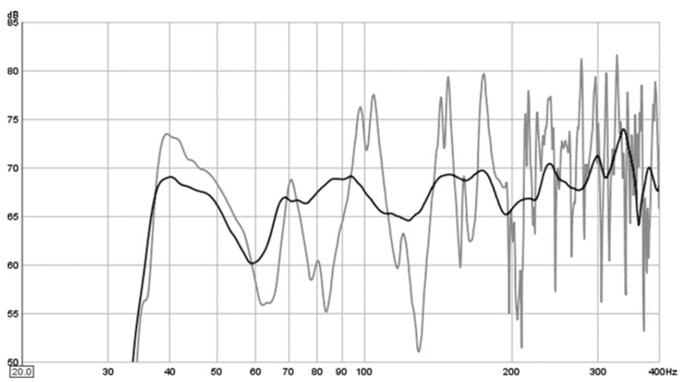

Figure 4: Adding bass traps flattens the low-frequency response of the room (darker line has bass traps) (Winer 459).

Loudspeaker Calibration

Testing in our acoustically treated environment would greatly flatten the frequency response of the audio system; however, the properties of the loudspeakers also contribute to spectral unevenness. Our team researched other methods to account for the frequency response of our loudspeakers.

Advanced methods

Ideally, our system would profile the acoustics of the entire room and calibrate our speakers accordingly. A few different methods exist to solve this problem. The first is the Finite Difference Time Domain algorithm used by Aslan and Paurobally in their study of active noise control systems. This algorithm models the sound source, room, and calibration microphone as mathematical models based on wave equations (Aslan and Paurobally 608). Another approach is to model sound as travelling along straight paths or “rays” and use the geometry of the room to create a model for the room’s acoustics (Siltanen 166). Both Aslan and Siltanen’s methods would provide a highly accurate model of the room environment, but they are far outside the scope of this project.

Our preferred method

We designed a much simpler calibration method for this project which would play a test tone and record the resulting output with a calibration microphone. We adapted our methodology from a patent from Intertrust Technologies Corproration which describes a system to correct spectral imbalances caused by the speakers and the room (Maher 1).

Roughly speaking, our test procedure would

1. Prompt the user to position a microphone near his/her ear
2. Initiate playback of a test tone (frequency sweep) from the speakers
3. Process resulting data and implement digital calibration filters (Maher 4)
4. Repeat until desired frequency response is achieved

A flowchart for our designed calibration routine is shown in Figure 5 below.

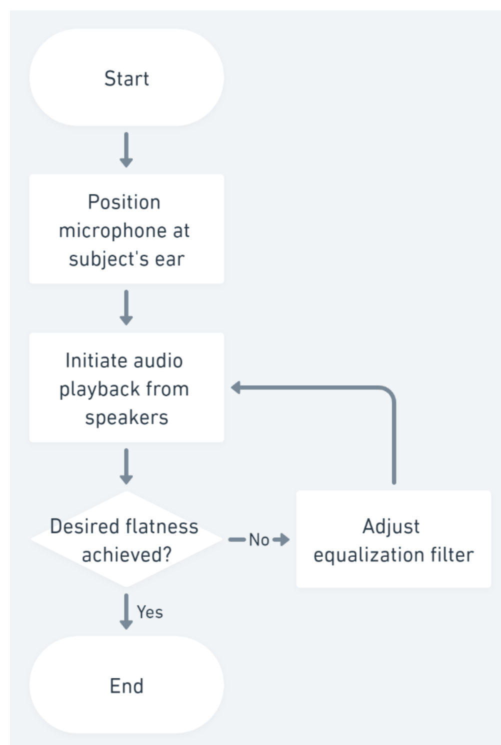

Figure 5: Our method of calibrating loudspeakers

The processing in step 3 compares the ideal, "flat" frequency response with the actual response measured by the calibration microphone and uses a digital filter to subtract out the difference (Maher 4). The patent recommends that the test tone include synchronization via some predefined pattern for more accurate processing (Maher 3). Another optional measure is to subtract the response of the microphone (Maher 4-5), but this step was unnecessary for our project because we use the same microphone every time we calibrate.

Conclusion

Unfortunately, our team was unable to complete the audio calibration portion of this project by the end of the semester. Unforeseen challenges and circumstances forced us to re-scope and use headphones in our final proof-of-concept, which did not require calibration for room acoustics. However, the methodology in this report will prove useful should our sponsor continue development on this project using loudspeakers. Further applications for this methodology include a home speaker setup or recording studio environment.

References

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